Listing of Claims:

This listing of claims will replace all prior versions and listings of claims in the application:

What is claimed is:

1. (currently amended) A method of modifying low frequency components of a digital audio signal having left and right channel signals, the method comprising the steps of: a) filtering the left and right channels signals using respective left and right high-pass filters to form left and right high-pass filtered signals; b) filtering the left and right channel signals using respective left and right band-pass filters to form left and right low frequency signals; c) modifying the amplitude of the left and right low frequency signals to give modified left and right low frequency signals whereby signals with amplitude a where $0 < a < a1$ are amplified by a first constant value C1, signals with amplitude a1 $\leq$ a < a2 are amplified proportional to 1/a, signals with amplitude a = ~~2a~~ a2 are unchanged, signals with amplitude a2 < a < a3 are attenuated proportional to 1/a, and signals with amplitude a =a3 are attenuated by a second constant value C2; and d) combining the modified band-pass filtered left and right signals with the respective left and right high-pass filtered signals to form respective modified left and right channel audio signals.

2. (original) A method according to claim 1 wherein in step c), the amplitude a of the signal is taken to be the amplitude of the left or right low frequency signal which has the largest absolute value.

3. (original) A method according to claim 2 wherein the first constant value C1 is 12.5.

4. (original) A method according to claim 1 wherein the second constant value C2 is 0.5.

5. (original) A method according to claim 1 wherein a1 = 0.04.

6. (original) A method according to claim 1 wherein a2 = 0.5.

7. (original) A method according to claim 1 wherein a3 = 1.

8. (original) A method according to claim 1 wherein the digital audio signal is an MP3 encoded signal.

9. (original) A method according to claim 1 wherein the digital audio signal is in WAV format.

10. (original) A method according to claim 1 wherein the parameters of the band-pass filters are user selectable.

11. (original) A method according to claim 1 wherein the parameters of the high-pass filters are user selectable.

12. (original) A method claimed in claim 1 using a limiter having a transfer function substantially as shown in FIG. 1d.

13. (Cancelled)

14. (new) An audio filtering system comprising at least one digital filter, the system configured to perform the method as recited in claim 1.

15. (new) The method as recited in claim 1 wherein the left and right band pass filters are implemented as Butterworth infinite impulse response filters.